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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/532,593	08/18/2005	Stuart Charles Wray	038665.56183US	4830

23911 7590 05/27/2009  
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EXAMINER
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CRUTCHFIELD, CHRISTOPHER M

ART UNIT	PAPER NUMBER
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2419

MAIL DATE	DELIVERY MODE
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05/27/2009

PAPER

**Please find below and/or attached an Office communication concerning this application or proceeding.**

The time period for reply, if any, is set in the attached communication.

<b>Office Action Summary</b>	<b>Application No.</b> 10/532,593	<b>Applicant(s)</b> WRAY ET AL.	
	<b>Examiner</b> Christopher Crutchfield	<b>Art Unit</b> 2419	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

### Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

### Status

- 1) ☒ Responsive to communication(s) filed on 04 May 2009.
- 2a) ☐ This action is **FINAL**.                      2b) ☒ This action is non-final.
- 3) ☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

### Disposition of Claims

- 4) ☒ Claim(s) 1-3 and 7-11 is/are pending in the application.
- 4a) Of the above claim(s) \_\_\_\_\_ is/are withdrawn from consideration.
- 5) ☐ Claim(s) \_\_\_\_\_ is/are allowed.
- 6) ☒ Claim(s) 1-3 and 7-11 is/are rejected.
- 7) ☐ Claim(s) \_\_\_\_\_ is/are objected to.
- 8) ☐ Claim(s) \_\_\_\_\_ are subject to restriction and/or election requirement.

### Application Papers

- 9) ☐ The specification is objected to by the Examiner.
- 10) ☐ The drawing(s) filed on \_\_\_\_\_ is/are: a) ☐ accepted or b) ☐ objected to by the Examiner.  
Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).  
Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).
- 11) ☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

### Priority under 35 U.S.C. § 119

- 12) ☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
- a) ☐ All    b) ☐ Some \*    c) ☐ None of:
1. ☐ Certified copies of the priority documents have been received.
  2. ☐ Certified copies of the priority documents have been received in Application No. \_\_\_\_\_.
  3. ☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

\* See the attached detailed Office action for a list of the certified copies not received.

### Attachment(s)

- |  |   |
|--|---|
| 1) <input checked="" type="checkbox"/> Notice of References Cited (PTO-892)          | 4) <input type="checkbox"/> Interview Summary (PTO-413)           |
| 2) <input type="checkbox"/> Notice of Draftsperson's Patent Drawing Review (PTO-948) | Paper No(s)/Mail Date. _____                                      |
| 3) <input type="checkbox"/> Information Disclosure Statement(s) (PTO/SB/08)          | 5) <input type="checkbox"/> Notice of Informal Patent Application |
| Paper No(s)/Mail Date _____  | 6) <input type="checkbox"/> Other: _____                          |

## **DETAILED ACTION**

### ***Transitional After Final Practice***

1. Since this application is eligible for the transitional procedure of 37 CFR 1.129(a), and the fee set forth in 37 CFR 1.17(r) has been timely paid, the finality of the previous Office action is hereby withdrawn pursuant to 37 CFR 1.129(a). Applicant's first submission after final filed on 4 May 2009 has been entered.

### ***Claim Rejections - 35 USC § 103***

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negated by the manner in which the invention was made.

3. The factual inquiries set forth in *Graham v. John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

1. Determining the scope and contents of the prior art.
2. Ascertaining the differences between the prior art and the claims at issue.
3. Resolving the level of ordinary skill in the pertinent art.
4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

4. This application currently names joint inventors. In considering patentability of the claims under 35 U.S.C. 103(a), the examiner presumes that the subject matter of the various claims was commonly owned at the time any inventions covered therein were made absent any

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evidence to the contrary. Applicant is advised of the obligation under 37 CFR 1.56 to point out the inventor and invention dates of each claim that was not commonly owned at the time a later invention was made in order for the examiner to consider the applicability of 35 U.S.C. 103(c) and potential 35 U.S.C. 102(e), (f) or (g) prior art under 35 U.S.C. 103(a).

5. **Claims 1 and 10** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Komatsu*, et al. (US Patent No. 6,914,900 B1) in view of *Kong*, et al. (Xiang Kong and Kenichi Mase, Dynamic Routing with Endpoint Admission Control for VoIP Networks, 2003 IEEE International Conference on Communications, 15 May 2003, Pages 1728-1732).

**Regarding claim 1**, *Komatsu* discloses a method of call admission control for a continuous stream of data in packet switched networks including at least two local area networks that communicate with one another across a connecting network, the method comprising determining a packet loss rate of previous calls to a second local area network and deciding to drop a call attempt based on the packet loss rate (Column 5, Lines 5-20, Column 7, Lines 15-27 and Column 3, Line 24). (The system of *Komatsu* discloses a system that maintains the packet loss rate from previous calls between two endpoints [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. The packet loss rates may be based on statistical analysis of the loss rates of multiple calls during a time period [Column 12, Lines 29-34]. When a call is made, the system references the aggregated loss statistics for that IP endpoint [i.e. local area network] from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call [Column 3, Line 24].)

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*Komatsu* fails to disclose a method further comprising deciding to determine a current packet loss rate based on the packet loss rate of previous calls, determining the current packet loss rate and deciding to drop a call based on the current packet loss rate. In the same field of endeavor, *Kong* discloses a method further comprising deciding to determine a current packet loss rate based on the packet loss rate of previous calls, determining the current packet loss rate and deciding to drop a call based on the current packet loss rate (Page 1728, Section II and Page 173, Section III.C). (The system of *Kong* discloses a system that probes multiple paths to determine available routes to a destination network [Pages 1728-1729, Sections II and III, Particularly Section III.B]. The system probes both a direct path [i.e. DR] between two networks and one of multiple indirect paths [i.e. PR's] between the source and destination calling node [Pages 1729-1730, Section III.B and Fig. 2]. During the probing process, the system determines the packet loss rate of the probes of both the direct [i.e. DR] and indirect [i.e. TR] paths [Pages 1728-1729, Section II and Fig. 1]. If the packet loss rate of the indirect [i.e. TR] path is too high, the path fails the check and is not selected for probing in a subsequent call [Page 1729, Section III.C]. However, if the packet loss rate of the path is acceptable, then the path is probed in future calls [Page 1729, Section III.C]. Therefore, the system of *Kong* decides to determine a packet loss rate for a network path only if the packet loss rate of a previous probe was acceptable.)

Therefore, since *Kong* suggests the use of previous packet drop rates to determine if a link should be probed and *Komatsu* discloses using the packet loss rate of previous calls to determine previous packet drop rates, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the conditional probing of *Kong* with the system of *Komatsu* by having the system of *Komatsu* probe connections to remote networks, but doing so only when the packet drop rates of prior calls on the connection are below a critical threshold. The motive to combine is to utilize active probing to provide up to date network statistics for

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admission control, while limiting probing bandwidth by not probing links which likely have unacceptable packet loss rates based on previous measurements.

**Regarding claim 10**, *Odom* fails to disclose a method wherein said step of determining a packet loss rate of previous calls comprises determining the packet loss rate from a first local area network to a second local area network. In the same field of endeavor, *Kong* discloses a method wherein said step of determining a packet loss rate of previous calls comprises determining the packet loss rate from a first local area network to a second local area network (Page 1728, Section II and Page 173, Section III.C). (The system of *Kong* discloses a system that probes multiple paths to determine available routes to a destination network [Pages 1728-1729, Sections II and III, Particularly Section III.B]. The system probes both a direct path [i.e. DR] between two networks and one of multiple indirect paths [i.e. PR's] between the source and destination calling node [Pages 1729-1730, Section III.B and Fig. 2]. During the probing process, the system determines the packet loss rate of the probes of both the direct [i.e. DR] and indirect [i.e. TR] paths [Pages 1728-1729, Section II and Fig. 1]. If the packet loss rate of the indirect [i.e. TR] path is too high, the path fails the check and is not selected for probing in a subsequent call [Page 1729, Section III.C]. However, if the packet loss rate of the path is acceptable, then the path is probed in future calls [Page 1729, Section III.C]. Therefore, the system of *Kong* decides to determine a packet loss rate for a network path only if the packet loss rate of a previous probe was acceptable.)

Therefore, since *Kong* suggests the use of previous packet drop rates to determine if a link should be probed and *Komatsu* discloses using the packet loss rate of previous calls to determine previous packet drop rates, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the conditional probing of *Kong* with the system of *Komatsu* by having the system of *Komatsu* probe connections to remote networks, but doing so

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only when the packet drop rates of prior calls on the connection are below a critical threshold.

The motive to combine is to utilize active probing to provide up to date network statistics for admission control, while limiting probing bandwidth by not probing links which likely have unacceptable packet loss rates based on previous measurements.

6. **Claim 2** is rejected under 35 U.S.C. 103(a) as being unpatentable over *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) in view of *Wing*, et al. (US Patent No. 7,496,044 B1).

**Regarding claim 2**, *Odom* discloses a method of call admission control for a continuous stream of data in packet switched networks (Page 1, Second Paragraph) including at least two local area networks (*Odom*, Page 4, Figure 4 and Fourth Paragraph) that communicate with one another across a connecting network (Page 4, Figure 4, WAN) the method comprising the steps of:

a. Determining current packet loss rate for calls from the first local area network to the second local area network (*Odom*, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [*Odom*, Figure 4] to the server gateway in the other network/LAN. It then measures the packet loss rate of reflected packets to determine the packet loss rate of calls between the two networks [*Odom*, Page 19, SAA Protocol and Calculated Planned Impairment Value]. This value, along with others is used to perform client access control.)

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b. Deciding to drop the call attempt based on the current packet loss rate (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (See Supra in [a])

c. Transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network (Odom, Page 19, SAA Protocol). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network [Odom, Page 19, SAA Protocol].)

d. Reflecting the burst of trial data received at the second node back to the first node (Odom, Page 19, SAA Protocol).

e. Receiving the reflected burst of trial data at the first node through the connecting network (Odom, Page 19, SAA Protocol).

f. Comparing the reflected burst of trial data to the transmitted burst of trial data to determine whether transmission of a continuous stream of data can be initiated from the first node in the first local area network to the second node in the second local area network (Odom, Page 19, SAA Protocol, Calculating Planned Impairment Value). (It is inherent that in order to determine packet loss in a ping style test [Odom, Page 18, SAA Probes Versus Pings], the reflected burst of trial data must be analyzed and compared to the data sent to determine if a portion of the burst was lost [i.e. if packet loss occurred].)



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g. The burst of trial data comprises a plurality of packets having a size that corresponds to packets that are to be sent if the call is completed (Page 19, SAA Protocol). (The SAA protocol may also send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set.)

*Odom* fails to disclose the first and second nodes comprise telephones and the burst of trial data comprises a plurality of packets having a size and priority that corresponds to packets that are to be sent if the call is completed. In the same field of endeavor, *Wing* discloses the first and second nodes comprise telephones and the burst of trial data comprises a plurality of packets having a size and priority that corresponds to packets that are to be sent if the call is completed (Column 9, Lines 31-53, Column 11, Lines 27-38 and Claim 1). (The system of *Wing* discloses a system that simulates a voice call before initiation by sending a bi-directional stream of real time protocol (RTP) no-op packets between the sending and receiving VOIP telephones [Column 9, Lines 31-53]. The packets may be the same size and have the same priority/class of service as the actual media packets that are to follow [Column 11, Lines 27-38 and Claim 1].)

Therefore, since *Wing* discloses the use of size and priority matching for call simulation between two telephone endpoints, it would have been obvious to combine the endpoint call simulation of *Wing* with the system of *Odom* by having the telephone endpoints transmit a trial burst of data with a priority and size that reflects the size and priority of the following voice packets, as taught by *Wing*, and reflecting the trial burst back to the sender, as taught by *Odom*. The motive to combine is to allow the endpoints to test the connection using a realistic probe packet, lowering the load on the call gateways and increasing the accuracy of the probing by

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matching the characteristics of the probe packet to the actual media packets that will make up the following voice call.

7. **Claims 3 and 11** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Odom* (*Odom*, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) in view of *Komatsu*, et al. (US Patent No. 6,914,900 B1), *Zuberi*, et al. (US Patent No. 7,366,097 B2) and *Wing*, et al. (US Patent No. 7,496,044 B1).

**Regarding claim 3**, *Odom* discloses a method of call admission control for a continuous stream of data in packet switched networks (Page 1, Second Paragraph) including at least two local area networks (*Odom*, Page 4, Figure 4 and Fourth Paragraph) that communicate with one another across a connecting network (Page 4, Figure 4, WAN) the method comprising:

a. Determining current packet loss rate for calls from the first local area network to the second local area network (*Odom*, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [*Odom*, Figure 4] to the server gateway in the other network. It then measures the packet loss rate to determine the packet loss rate of calls between the two networks. This value, along with others is used to perform client access control.)

b. Wherein said step of determining a current packet loss rate comprises transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network, reflecting the burst of trial

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data received at the second node back to the first node, receiving the reflected burst of trial data at the first node through the connecting network and comparing the reflected burst of trial data to the transmitted burst of trial data to determine whether transmission of a continuous stream of data can be initiated from the first node in the first local area network to the second node in the second local area network (Odom, Page 19, SAA Protocol). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network which reflects the data packets back to the sending gateway for analysis [Odom, Page 19, SAA Protocol].)

c. Wherein the burst of trial data comprises a plurality of packets having a size that corresponds to packets that are to be sent if the call is completed (Page 19, SAA Protocol). (The SAA protocol may also send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set.)

d. Deciding to drop a call attempt based on the last measured packet loss rate and historic packet loss rates for probes on the connection (Pages 22-24). (The system of *Odom* discloses that when a destination endpoint is cached, the system may periodically probe the endpoint [Pages 22-24]. Connections are then admitted or denied based on both the results of the most current probe and the previous probe [Pages 22-24].)

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*Odom* fails to disclose a method further comprising determining the packet loss rate based on data collected from calls and not data probes. In the same field of endeavor, *Komatsu* discloses a method further comprising determining the packet loss rate based on data collected from calls and not data probes (Column 5, Lines 5-20, Column 7, Lines 15-27 and Column 3, Line 24). (The system of *Komatsu* discloses a system that maintains the packet loss rate from a previous calls between two endpoints. [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the system references the aggregated loss statistics for that IP endpoint [i.e. local network] from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call [Column 3, Line 24].)

Therefore, since *Komatsu* discloses call admission based on previous call performance, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the historic admission control of *Komatsu* into the teachings of *Odom*. The historic admission control of *Komatsu* can be combined with the system of *Odom* by having the system of *Odom* record connection statistics in its cache, as taught by *Komatsu*, and perform admission control on incoming calls based on current and past connection statistics, as taught by *Odom*. The motive to combine is provided by *Komatsu* and is to reduce the delay in setting up a new call (Column 5, Lines 5-20).

*Odom* as modified by *Komatsu* fails to disclose a method comprising determining the success rate of previous calls from a first local area network to a second local area network and deciding to drop the call attempt based on the current packet loss rate and the success rate of previous calls. In the same field of endeavor, *Zuberi* discloses a method comprising determining a method comprising determining the success rate of previous calls from a first local area network to a second local area network and deciding to drop the call attempt based on the

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current packet loss rate and the success rate of previous calls (Column 15, Lines 2-22 and Column 13, Line 30 to Column 14, Line 56). (The system of *Zuberi* discloses a bandwidth assessment probe for assessing bandwidth for transmitting audio [Column 13, Line 30 to Column 14, Line 56 and Column 7, Lines 8-27]. *Zuberi* further discloses that the admission success or rejection of previous connections on the same link is stored in an admission control cache [Column 13, Line 30 to Column 14, Line 56]. When a new connection is made with the system, the system checks the admission control cache, if a previous session on the link was admitted, then the system performs only an abbreviated probe of the stream [Column 15, Lines 2-22 and Column 14, Lines 20-30].)

Therefore, since *Zuberi* discloses using past admission success and present probe results in admitting connections and *Odom* as modified by *Komatsu* discloses admitting calls based on the past and present network conditions, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the success based probing of *Zuberi* into the system of *Odom* by having the system of *Odom* store the success or failure of previous calls to an endpoint in its cache and to then verify a connection to a cached endpoint using a probe, as taught by *Zuberi*. The motive to combine is provided by *Zuberi* and is to reduce probing traffic connection setup time by requiring only a brief validation of the connection using a shortened probe (Column 15, Lines 2-22).

*Odom* fails to disclose a method wherein the burst of trial data comprises a plurality of packets having a size and priority that corresponds to packets that are to be sent if the call is completed. In the same field of endeavor, *Wing* discloses a method wherein the burst of trial data comprises a plurality of packets having a size and priority that corresponds to packets that are to be sent if the call is completed (Column 9, Lines 31-53, Column 11, Lines 27-38 and Claim 1). (The system of *Wing* discloses a system that simulates a voice call before initiation by

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sending a bi-directional stream of real time protocol (RTP) no-op packets between the sending and receiving VOIP telephones [Column 9, Lines 31-53]. The packets may be the same size and have the same priority/class of service as the actual media packets that are to follow [Column 11, Lines 27-38 and Claim 1].)

Therefore, since *Wing* discloses the use of size and priority matching for call simulation between two endpoints, it would have been obvious to combine the endpoint call simulation of *Wing* with the system of *Odom* by having the endpoints transmit a trial burst of data with a priority and size that reflects the size and priority of the following voice packets, as taught by *Wing* and reflecting the trial burst back to the sender, as taught by *Odom*. The motive to combine is to allow the endpoints to test the connection using a realistic probe packet, thereby increasing the accuracy of the probing by matching the characteristics of the probe packet to the actual media packets that will make up the following voice call.

**Regarding claim 11**, *Odom* fails to disclose a method wherein the first node comprises a telephone in the first local area network and the second node comprises a telephone in the second network. In the same field of endeavor, *Wing* discloses a method wherein the first node comprises a telephone in the first local area network and the second node comprises a telephone in the second network (Column 9, Lines 31-53, Column 11, Lines 27-38 and Claim 1). (The system of *Wing* discloses a system that simulates a voice call before initiation by sending a bi-directional stream of real time protocol (RTP) no-op packets between the sending and receiving VOIP telephones [Column 9, Lines 31-53].)

Therefore, since *Wing* discloses call simulation between two telephone endpoints, it would have been obvious to combine the endpoint call simulation of *Wing* with the system of *Odom* by having the telephone endpoints transmit a trial burst of data, as taught by *Wing* and reflecting the trial burst back to the sender, as taught by *Odom*. The motive to combine is to

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allow the endpoints to test the connection using the endpoints as opposed to the gateway, thereby reducing the load on the gateway.

8. **Claim 7** is rejected under 35 U.S.C. 103(a) as being unpatentable over *Komatsu*, et al. (US Patent No. 6,914,900 B1) and *Kong*, et al. (Xiang Kong and Kenichi Mase, Dynamic Routing with Endpoint Admission Control for VoIP Networks, 2003 IEEE International Conference on Communications, 15 May 2003, Pages 1728-1732) as applied to claim 1 and further in view of *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26).

**Regarding claim 7**, *Komatsu* fails to disclose a method wherein said step of determining said current packet loss rate comprises transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network, reflecting the burst of trial data received at the second node back to the first node, and receiving the reflected burst of trial data at the first node through the connecting network. In the same field of endeavor, *Odom* discloses a method according wherein said step of determining said current packet loss rate comprises transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network, reflecting the burst of trial data received at the second node back to the first node, and receiving the reflected burst of trial data at the first node through the connecting network (Odom, Page 19, SAA Protocol). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network which reflects the packets back to the sender to be analyzed [Odom, Page 19, SAA Protocol].)

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Therefore, since *Odom* disclose the use of reflected trial data, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the reflected probes of *Odom* into the teachings of *Komatsu* as modified by *Kong* by reflecting probe packets. The motive to combine is to test the connection bi-directionally without having to generate a second set of test packets at the receiving node.

9. **Claims 8 and 9** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Komatsu*, et al. (US Patent No. 6,914,900 B1) and *Kong*, et al. (Xiang Kong and Kenichi Mase, Dynamic Routing with Endpoint Admission Control for VoIP Networks, 2003 IEEE International Conference on Communications, 15 May 2003, Pages 1728-1732) as applied to claim 1 and further in view of *Wing*, et al. (US Patent No. 7,496,044 B1).

**Regarding claim 8**, *Komatsu* fails to disclose a method wherein the first node comprises a telephone and said second node comprises a telephone. In the same field of endeavor, *Wing* discloses a method wherein the first node comprises a telephone and said second node comprises a telephone (Column 9, Lines 31-53, Column 11, Lines 27-38 and Claim 1). (The system of *Wing* discloses a system that simulates a voice call before initiation by sending a bi-directional stream of real time protocol [RTP] no-op packets between the sending and receiving VOIP telephones [Column 9, Lines 31-53].)

Therefore, since *Wing* discloses call simulation between two telephone endpoints, it would have been obvious to combine the endpoint call simulation of *Wing* with the system of *Odom* by having the telephone endpoints transmit a trial burst of data, as taught by *Wing* and reflecting the trial burst back to the sender, as taught by *Odom*. The motive to combine is to



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allow the endpoints to test the connection using the endpoints as opposed to the gateway, thereby reducing the load on the gateway.

**Regarding claim 9**, *Komatsu* fails to disclose a method wherein the burst of trial data comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is completed. In the same field of endeavor, *Wing* discloses a method wherein the burst of trial data comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is completed (Column 9, Lines 31-53, Column 11, Lines 27-38 and Claim 1). (The system of *Wing* discloses a system that simulates a voice call before initiation by sending a bi-directional stream of real time protocol (RTP) no-op packets between the sending and receiving VOIP telephones [Column 9, Lines 31-53]. The packets may be the same size and have the same priority/class of service as the actual media packets that are to follow [Column 11, Lines 27-38 and Claim 1].)

Therefore, since *Wing* discloses the use of size and priority matching for call simulation between two endpoints, it would have been obvious to combine the endpoint call simulation of *Wing* with the system of *Odom* by having the endpoints transmit a trial burst of data with a priority and size that reflects the size and priority of the following voice packets, as taught by *Wing* and reflecting the trial burst back to the sender, as taught by *Odom*. The motive to combine is to allow the endpoints to test the connection using a realistic probe packet, thereby increasing the accuracy of the probing by matching the characteristics of the probe packet to the actual media packets that will make up the following voice call.

***Response to Arguments***

10. Applicant's arguments with respect to claims 1-3 and 7-11 have been considered but are moot in view of the new ground(s) of rejection.

***Conclusion***

11. The following prior art made of record and not relied upon is considered pertinent to applicant's disclosure:

a. *Shah*, et al. (US Pre Grant Publication No. 2005/0027861) - Giving additional information on a system similar to that of *Odam*.

b. *Murphy*, et al. (US Patent No. 6,542,499) – Disclosing a system that monitors VoIP link quality and re-directs calls to a PSTN if the quality falls below a pre-defined threshold.

c. *Tuomi*, et al. (US Pre Grant Publication No. 2002/0110112 A1) – Disclosing a VoIP system that determines the availability of multiple endpoints based on continuous periodic polling.

d. *Bonadaro*, et al. (Patent Application No. 2006/0239204 A1) – Disclosing a VoIP admission control system that periodically probes remote networks.

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e. *Kelly*, et al. (Patent Application No. 2004/0174815 A1) - Disclosing an endpoint admission control system using passive (i.e. network sniffing) and active (i.e. probing) determination of network conditions.

f. *Mase*, et al. (Kenichi Mase and Yuichiro Toyama, End-to-End Measurement Based Admission Control for VoIP Networks, 2002 IEEE international Conference on Communications, 2002, Pages 1194-1198) – Disclosing an admission control algorithm that uses continuous or request based probing to determine availability of remote VoIP networks.

g. *Ivars*, et al. (Ignacio Ivars and Gunnar Karlsson, PBAC: Probe-Based Admission Control, Springer Lecture Notes in Computer Science, 2001, Pages 97-109) – Disclosing an endpoint based admission control scheme that uses probing and packet drop counting to admit/deny connections wherein denied connections are subject to a back off period before attempting to reconnect.

h. *Bilhaj*, et al. (Abdulkhalig Bilhaj and Kenichi Mase, Endpoint Admission Control Enhanced Systems for VoIP Networks, Proceedings of the 2004 International Symposium on Applications and the Internet, Pages 1-4) – Disclosing another probe based admission control system based on current and historic packet loss and delay.

i. *Bianchi*, et al. (Giuseppe Bianchi and Nicola Blefari-Melazzi, Admission Control Over Assured Forwarding PHBs: A Way to Provide Service Accuracy in a DiffServ

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j. *Kelly* (Tom Kelly, An ECN Probe-Based Connection Acceptance Control, ACM SIGCOMM Computer Communications Review, Issue 3, 2001, Pages 14-25) –

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k. *Breslau*, et al. (Lee Breslau, Edward Knightly, Scott Shenker, Ion Stoica and Hui Zhang, Endpoint admission control: architectural issues and performance, ACM SIGCOMM, 2000, Pages 57-69) – Disclosing a comparison of multiple endpoint admission control algorithms for probe-based admission control.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Christopher Crutchfield whose telephone number is (571) 270-3989. The examiner can normally be reached on Monday through Friday 8:00 AM to 5:00 PM EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Daniel Ryman can be reached on (571) 272-3152. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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